
F1
F2
F3

LINEAR ACOUSTIC

Viewers are listening.

That's what it really comes down to.

The advent of digital may have brought whiplash transformation to television broadcast engineering but something else changed along the way - audience expectations.

Viewers are more discerning.

Linear Acoustic products and technologies are designed to manage audio and loudness, help you maintain compliance, upmix, downmix, encode, decode, meter, monitor every audio function along the path from production to transmission intuitively.

But our primary goal is audio quality. Our focus remains on helping broadcasters world wide deliver compelling, engaging audio that is naturally compliant because it satisfies viewers.

Viewers are listening.

Christina Carroll
SVP, Global Sales, Telos Alliance

LINEAR ACOUSTIC
COMPANY HISTORY

AUDIO UNDER CONTROL

Audio has been my passion for as long as I can remember, and growing up in the NY area fed nicely into this desire. From repairing headphones in grade school (long before truly understanding what I was doing, and only occasionally bringing them back to life), to being the chief engineer of my high school and college radio stations. It was in radio that I developed an insatiable curiosity about broadcast and audio processing in particular. Thankfully, I had several very patient teachers that grimaced and looked the other way when I was "understanding" stuff using the disassembly method of learning. Some of it actually made it back together, and in hindsight, luckily some of it did not.

TO DOLBY AND BEYOND

Joining Dolby Laboratories in 1995 was a seminal event. There was not a broadcast group per se, but the market for digital sound on film was growing and engineers were needed. Spending almost four years on the film stages of New York delivered in-depth experience with the production of matrix and discrete surround audio and with the brand new DVD format which specified the Dolby Digital (AC-3) codec for multichannel audio.

Almost simultaneously, the AC-3 codec was mandated for use in the ATSC digital television standard, and soon thereafter was included in the DVB specification. The rush was on to develop the tools for this new format.

Being walking distance from the major television networks in New York allowed me to experiment with early DTV audio products in some of the worlds most advanced broadcast facilities.

Next stop was Dolby's San Francisco headquarters to take on the role of the professional audio product manager. Here a rogue team of engineers and coding experts developed a set of products that laid the foundation for broadcasters to transition to digital video and audio and from mono or stereo to 5.1 channel surround. Never in the history of the company had so many products been developed in so short a time, but it was necessary as we were not just adding more channels but changing the entire path from production to consumer. The dream was that Hollywood audio quality could be delivered to the home via broadcast, and for the first time transmission methods would not get in the way.

There were two problems with this dream. First, it required everything to be in place all at once to make it work seamlessly. This might be practical in the lab but reality dictates it will work out otherwise. Second, there seemed to be so many places where things could go wrong and it was becoming very obvious that some sort of overall protection was necessary.

The time had come to take my bag of collected tricks and hit the road as a consultant to try to help broadcasters and manufacturers take the next steps. However, it quickly became apparent that the technology to avert a potential train wreck would have to be homegrown.

YEP, WE STARTED IN A NJ GARAGE

Romantic, isn't it? Actually, Linear Acoustic was started in the basement of the house I grew up in and expanded into the garage (and the dining room, living room, and at least one bedroom). It also consumed a good deal of space at Leif Claesson's house in California where he turned our good ideas into algorithms. Interestingly, we were never on the same coast during the entire development but overnight delivery service and the Internet made us feel like we were in the garage together.

Once we finished the initial development of the first Linear Acoustic product called the OCTiMAX 5.1, Leif and I showed it at the SMPTE convention in Pasadena, CA. Thankfully, we caught someone's attention.

That someone was Steve Smith, the venerable engineering leader of Liberty Corporation and he was tasked with transitioning his television stations to digital. When we informed him that we were working on a DTV loudness controller, he proposed that if we could make digital television audio as hands-off as it was in analogue and still preserve the quality that he would outfit all sixteen of his television stations. Steve and Liberty became our first customer.

We were working on a shoestring budget creating products that were being installed by some of the top US television broadcasters as they began their transition to digital. Every unit was hand-assembled and carefully tested using tools that are common today, but were new to the industry then. As with any brand new product, there were bugs, but most of ours had four or more legs and were removed with compressed air.

AND THEN WE MOVED TO PA

Soon we outgrew the garage and moved to Lancaster, Pennsylvania to enable us to bring on some additional engineering talent and to take advantage of easier access to better quality high-tech manufacturing vendors.

In Lancaster, we began R&D that resulted in the first ever AC-3 splicer (and they said it couldn't be done), along with a higher density audio transport system called e-squared was used on such high profile events as the Academy Awards and the Country Music Awards broadcasts. We also innovated some metadata tools and a really slick audio and metadata monitoring system.

In 2008, we proudly joined Steve, Frank and Mike to become part of the Telos Alliance.

Today, the end of analogue over the air television is a reality in the US and is in process internationally, and loudness problems are rampant. Sadly, this was predicted.

Thankfully, we are amidst the continuing release of new and useful products that are the culmination of our work since the beginning of Linear Acoustic. Our approach supports industry efforts to solve loudness problems by working on each stage of the chain rather than just slapping a "cruncher" at the end. We also have the leading stereo to 5.1 channel upmixing technology called UPMAX and have recently released a new flavor called UPMAX II to critical acclaim. New software tools allow for the most advanced file-based audio processing available.

AND THEN WE WON AN EMMY

We are incredibly humbled and honored to have been presented with a 2010 Technical Emmy® award for "The Pioneering Development of an Audio/Metadata Processor for Conforming Audio to ATSC Standard" (whew).

We take this honor very seriously and recognize the importance of remaining active participants in standards creation within the ATSC for DTV and Mobile DTV, SMPTE, and the EBU to continue to make audio even better.

In addition to having some of the best ears in the industry for broadcast audio, our ears are also sensitive to your feedback and suggestions. Our products are based on direct suggestions (or commands) from customers.

Broadcast is in our blood: it is what we do, it is what we love to do. It is what links us to you, our customers and our colleagues. Viewers ARE listening, and so are we.

TIM CARROLL
FOUNDER AND PRESIDENT, LINEAR ACOUSTIC

DELIVERING QUALITY SOUND

A FAREWELL TO LOUDNESS

Thank you, loudness. Thank you for your spikes, sudden bursts and consistent inconsistencies. Thank you for transforming serene, satisfied television viewers into an angry, ear-cupping citizenry, banding together to complain to their governments. But most of all, thank you for finally causing the broadcast industry to bolt upright and recognize a seemingly obvious truth: TV is more than simply a picture. And for that, loudness deserves some gratitude.

By reminding TV broadcasters of the importance of audio, loudness, in a way, has made TV quality better than it might have been.

When broadcast engineers focus on the real audio issue – delivering consistent quality sound – loudness becomes moot, consumers are contented, regulators draw back, producers can be satisfied, and broadcasters are happy.

It is also important to remember that while it is easy to blame loudness issues on commercial advertisements or other interstitials, it can also be the fault of the programs themselves. A train crash and explosion might be fine during a matinee but it is not going to go over very well at 3 in the morning with kids asleep.

PLEASE MORE PEOPLE, MORE OF THE TIME

Everyone wants your audio signal to be perfect, at least to their expectations. Who requires it the most? Is it the regulator? The station manager? The program producer? The consumer? In reality, it is all of the above but for different and sometimes opposing reasons.

Each of these targets has their own benchmark for satisfaction. The station wants happy viewers and a happy regulator. The regulator wants no complaints from viewers.

Ultimately then, the final judge is the viewer. It is the viewers who create station ratings and thus a place for paid advertising to be shown which generates revenue to buy programming and pay staff. It is the same viewers who will complain to the regulator when the audio is not right - especially when there are unexpected loudness shifts.

The viewer wants consistent audio, then they will not complain to the regulator and the station is likely safe. How this is accomplished, however, may not satisfy the program producer.

COMPLIANCE OR QUALITY?

Is it possible to regulate an audio signal to the point of being unlistenable? For some governments, nothing is impossible. Although regulators may specify both a loudness target and the method for measuring loudness, they will likely not react unless they receive viewer complaints. However, a regulated target and a metering specification, if approached blindly, may result only in the meter being happy.

Just like in the departed or soon to be departed days of analogue, devices can be installed at the end of the chain prior to transmission that raises or lowers gain depending on how much the loudness of the incoming audio varies from the target. This is commonly referred to as Automatic Gain Control (AGC). AGC systems will more or less treat every shift similarly and will affect the good and the bad. Everything gets a little something whether it needs it or not.

While there are many sophisticated (and some unsophisticated ways) to accomplish AGC, in reality there is no way for any machine, regardless of manufacturer, topology, or promise of magic outcome that can, in real time, know the difference between a good, intentional loudness shift and a bad, annoying loudness shift. Certainly human generated commands and even metadata can be used to change or bypass processing for content that is believed to be good, but this involves a great deal of effort that has proven thus far not to have been exerted. Television mixers have long been used to the idea that what was transmitted via analogue means would be different than what they created and that was just the way it was. In today's digital world, there is no technical reason why before and after cannot match. It happens when films mixed for the big screen are then transferred to DVD and the same audio coding system is used for 5.1 channel broadcasts.

TAKE IT IN DOSES

Since machines cannot automatically know the difference, a better approach is to separate the problem into smaller tasks: matching average loudness, managing transitions, and delivering appropriate dynamic range.

Taken separately, a much better result can be obtained. For example, to match the average loudness of programs, use of BS.1770 along with either manual or automatic control of the meter based on an anchor element such as dialogue, allows the overall average to be measured and then a simple overall one-time level scale to be applied so the target is achieved. This changes nothing about the content and preserves the intent of the producer.

Matching average loudness of different pieces of content does not, however, solve jarring transitions. These occur at program boundaries and are the result of a mismatch of short term loudness at the junction between the end of one piece of content and the beginning of the next.

Think of a program whose average loudness measures at one level and a commercial advert that measures at another level. If the averages are matched by scaling their loudness, then on average they will sound equally loud. However, if a dramatic program is ending with a quiet death scene (as they often do), and compared to average is quiet for the 60 seconds leading up to the advert, guess what is about to happen? Yep, the advert will seem too loud. Guess what else? Meters will be totally happy since they are looking at longer term averages.

To oversimplify things, this is in fact similar to a dynamic range issue. It is not the same dynamic range variation as a loud train crash or gun fight in an action adventure movie, which is expected, but is instead an artificial variation. In fact the difference in loudness in this case may be much less than the gun fight. However, it is perceived as much worse. Likely this is because disparate elements are being glued together not for artistic reasons, but for financial reasons. The commercial must play at a certain time, per contract, whether or not it matches the program that surrounds it. How can this be captured by any meter? So far, it cannot be, at least not in real time.

One way to fix this is to use the AGC techniques described earlier which to minimize variations. Again, this will keep the viewer and the meter happy, but the program will have been irreparably changed and the producers will probably not be thrilled with the outcome.

The other way is to capture it in the program delivery specification. Offering typical ranges for programming and examples of what might happen at program/commercial boundaries will enable mixers to take control of the situation and make better artistic choices than any machine could make.

It is also worthwhile to refer mixers and program producers to recommended practices that offer guidance on speaker calibration in mix environments. Interestingly, monitoring at levels closer to what a typical consumer might listen at results in mixes with more appropriate dynamic range. Since the difference between average loudness and background noise in the mix room is now smaller, the dynamic range of the program must also be reduced.

SUMMARY

The intention of regulators is to satisfy their human constituents. Meters and loudness targets are well intentioned but when relied upon as the sole arbiter of compliance often lead to content that while consistent may be overly so. Like gravy without the occasional lump, the excitement of variance is gone. The trick is to preserve the good variance and manage the not so good variance.

To make this work requires more effort from broadcasters and program producers. Absent metadata systems to manage all of this, broadcasters must supply accurate and achievable program delivery specifications and producers must take into account the typical viewer and what may or may not be appropriate to deliver to them. Since broadcasters may be legally required to satisfy viewers, if content does not fit, they may be forced to make it fit. If both sides know the rules and have goals that are realistic and mostly aligned, it is possible to achieve acceptable balance.

The guidelines for making it all work boil down to one rule we should all post on our walls: don’t upset the viewer stupid! This is where it all begins and ends. No complaints=happy regulator=happy broadcaster. The challenge then is to manage the cost of this satisfaction versus preservation of content. It can be done.

If not, there are always AGCs that can smooth out everything. Everything.

Consistent quality sound, delivered with perfectly tuned images. This is what makes great television for program producers, regulators, consumers and ultimately, broadcasters.

And a final hat tip to loudness, without whom we may never have made it to this point.

TIM CARROLL
FOUNDER AND PRESIDENT, LINEAR ACOUSTIC

AERO.air

AUDIO/LOUDNESS MANAGER

Loudness Control, Upmixing, AES and SDI I/O plus Optional Dolby® Encoding and Decoding, Nielsen, and Compensating HD/SD-SDI Video Delay

AERO.air® is audio purity for digital television.

It provides proven loudness control, decoding and encoding, and unmatched upmixing capabilities. Factory presets ensure audio quality and easy set-up, while experienced users will appreciate extensive access to individual controls. Adjust the AERO.air for wideband multi-stage processing, multiband multi-stage processing, or anywhere in between.

AERO.air accepts 5.1-channel and two-channel audio via included AES or HD/SD-SDI inputs, plus dedicated EAS/Aux bypass inputs. Audio is then processed by the multiband, multistage ITU-R compliant AEROMAX® loudness cores resulting in smooth audio with appropriate dynamics. Two-channel audio is automatically upmixed producing a consistent surround-field, perfectly downmix compatible for all stereo viewers.

If present, audio metadata will manage upmixing and improve loudness control while minimizing impact on source audio. Extensive fallback options enable the AERO.air to compensate for missing or incorrect metadata.

Industry standard two-channel to 5.1-channel upmixing is provided by the Hollywood approved UPMAX® and UPMAX II algorithms. AutoMAX-II provides automatic and GPI or metadata guided control of upmixing without risking loss of center channel dialogue.

A fully processed selectable LtRt surround or LoRo stereo downmix of the main program audio is provided at all times for legacy stereo distribution paths or simple local monitoring. A 6.3mm (1/4”) high-level headphone connector and VGA output for multi-screen display complete the package.

Extensive standard I/O includes up to ten main AES inputs and outputs and front panel headphone connector. HD/SD-SDI I/O, with or without compensating video delay, enables de-embedding and re-embedding up to 16 channels of audio plus SMPTE 2020 (VANC) metadata. All AES outputs remain active when SDI option is enabled. Embedded channels can be routed through or around processing. Encoded signals can be de-embedded and re-embedded.

A bright color display, large rotary encoder, and four control keys provide simple menu navigation and adjustment. The AERO.air can be controlled remotely via GPI/O, while Gigabit Ethernet allows TCP control by automation systems.

The AERO.air contains dual redundant power supplies and hard relay bypass for the digital audio, SDI, and metadata signals, necessary in mission-critical applications.

Software options can generally be added in the field. Hardware options such as Dolby encoding or decoding and video delay must be factory installed.

OPTIONS

NIELSEN OPTION:

Generates revenue critical NAES II and Nielsen Watermark audience measurement codes. AERO.air precisely inserts these signals for maximum code recovery – after audio decoding and processing and before transmission encoding.

DOLBY DECODING OPTION:

Allows reference quality decoding of Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), and Dolby E content from any AES or SDI input signal.

DOLBY ENCODING OPTION:

Two Dolby Digital (AC-3) and/or Dolby Digital Plus (E-AC-3) encoders for 5.1 plus stereo audio.

AERO.AIR IS AVAILABLE IN TWO VERSIONS:

AERO.AIR (DTV):

AERO.air (DTV): Provides 5.1 channel loudness control and upmixing and outputs full-time 5.1 plus a stereo downmix.

AERO.AIR (5.1):

AERO.air (5.1): Supports full 10-channel 5.1 + 2 + 2 and 5 x stereo modes, and includes dual upmixers and CrowdControl™ dialogue protection.

AERO.one

AUDIO/LOUDNESS MANAGER

Audio/Loudness Manager for Stereo, 5.1 and 16 Channel Audio HD/SD-SDI I/O and optional Dolby® Digital (AC-3)/Dolby Digital Plus Encoding

AERO.one® is audio perfection for digital television.

Built in loudness control, metadata control, and optional transmission encoding make AERO.one the ideal choice for stations that want to provide a seamless, optimum quality experience for their audience. Now your viewers can be protected from loudness shifts and loss of surround sound in a simple, cost effective, compact, and feature rich manner.

AERO.one is well suited for main or backup transmission paths, providing high quality audio in a feature rich and cost effective manner.

Like other processors in the Linear Acoustic AERO family, the AERO.one accepts up to eight pairs of PCM audio (4 pairs via AES, up to eight pairs via SDI) to handle from two channel up to dual 5.1+2 channel audio programming. The unit can apply adaptive wideband and multiband, multistage ITU compliant loudness control and upmixing to the audio, with or without metadata guidance, to tame loudness and image shifts while preserving more of the original content than previously possible.

Upmixing is provided by the Hollywood-approved UPMAX® algorithm which provides a compelling 5.1-channel Audio experience while remaining completely downmix compatible. AERO.one includes the AutoMAX-II auto-detection algorithm to smoothly and automatically bypass upmixing when 5.1-channel audio is applied.

Upmixing and processing modes can be controlled by a combination of GPI contact closures and applied metadata.

Downmixed versions of the main programs are available at all times. These signals can be either a stereo LoRo downmix or an industry standard LtRt surround encoded mix.

Metadata can be applied, if available, via the VANC space of an applied HD-SDI signal or from a standard serial input to control of upmixing and processing functions. Extensive protection is provided to prevent audible effects of incorrect or missing metadata.

A highly-visible LED display and simple navigation cluster provide easy function adjustment. Relay bypass of all signals for trouble-free operation in transmission critical environments.

Available options include internal 5.1 channel Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) Encoding and SNMP monitoring.

AERO.ONE IS AVAILABLE IN FOUR VERSIONS:

AERO.ONE (16) – 2+2+2+2 through 5.1+2+5.1+2 channel loudness control plus quad UPMAX upmixing engines and dual downmixed outputs.

AERO.ONE (V3) – 5.1+2 channel loudness control plus dual UPMAX upmixing engines and downmixed output.

AERO.ONE (DTV) – 5.1 channel loudness control plus upmixing and downmixed output.

AERO.ONE (TV) – 2+2 dual stereo programs plus SAP and CrowdControl™. CrowdControl dialogue protection is vital for sports broadcasts, ending the annoying dialogue loss that occurs in mixes rich in sound effects.

AERO.lite

AUDIO/LOUDNESS MANAGER

Cost-effective 2-channel CALM and R128 loudness compliance - that's what AERO.lite™ delivers. Solve inconsistent audio loudness with a simple set-and-forget, feature-rich stereo processor that incorporates award-winning loudness control tools and extensive I/O. Perfect for main or backup transmission paths.

Input and output signal levels are displayed alongside processing meters, and the optional ITU-R BS.1770 measured LKFS output loudness value is provided to give instant verification of loudness compliance.

Audio can be extracted from any pair of an applied HD/SD-SDI signal, AES, or balanced analogue inputs and routed to the processing core. Output is provided simultaneously via the front panel 6.3mm (1/4") headphone connector, +4dBu balanced analogue outputs, AES output and for re-embedding into any or all SDI pairs. Analogue or AES inputs can be used as sources for embedding even if not used for processing. Since all 16 channels are available for de-embedding and re-embedding, pair shuffling can be easily accomplished.

Designed and assembled in the USA, the lightweight and rugged 1RU aluminum chassis is durable enough for installation in challenging environments like OB trucks and cramped edit bays. Built on a broadcast quality Linear Acoustic platform, the AERO.lite is professional grade equipment.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. Failover bypass relays on all I/O maintains signal continuity. Auto-ranging power supply for worldwide compatibility, and sealed, locking 2.5mm DC input connector for available redundant power supply.

AERO.1000

AUDIO/LOUDNESS PLATFORM

High Density Metadata Based Loudness Control with Dolby® Coding Processing, Upmixing, ITU-R BS.1770 Metering, TCP/IP Remote Control, and Redundant PSU

» Fully-Featured TCP/IP Remote Control Application
» HTTP and Web Server included

AERO.1000™ is a fresh, revolutionary approach to balancing density, control, and quality. Award-winning loudness control tools plus extensive I/O in a flexible, expandable, high density package make the AERO.1000 a wise investment.

» Up to 8 AEROMAX® audio engines including UPMAX® » Linear Acoustic CARBON™ Hybrid Processing » Up to 8 Dolby decoders and 8 Dolby Encoders » Utility ITU-R BS.1770 loudness meters » 3GHz HD/SD-SDI I/O with compensating video delay » Up to 16 channels of AES I/O with reference input » Stereo +4dBu Analogue I/O » Front panel headphone output » TCP/ IP remote control and HTTP server » Redundant PSU » Up to 8 Nielsen watermark encoders » DVB-ASI I/O (optional)

VIEWERS ARE LISTENING. METERS ARE METERING.

Now that the industry is focused on loudness, solutions are rampant. Unfortunately, sound quality is mostly forgotten in favor of meter satisfaction. Linear Acoustic has innovated and ardently supports the approach of getting loudness matched to target upstream using metering and/or file-based scaling tools. Leave the final polishing provided by dynamic range control (DRC) as a part of transmission.

Linear Acoustic CARBON™ Hybrid Processing is a patent-pending hybrid between multiband techniques and metadata. Because it is created during transmission encoding, this metadata requires no operator intervention or special tools - it is a new version of the DRC part of the Dolby Digital encoder that has always been there. Except it is now effective and uncomplicated.

Applied by default in all consumer decoders, metadata provides DRC that can be disabled in higher-end systems. How does it sound? Exactly like the high-quality audio control always provided by Linear Acoustic: Exceptional. The difference is that the audio can remain untouched. Or not. Broadcasters can choose permanent control where necessary, leaving reversible control for high quality trusted programming.

Handling up to 64 channels of audio, encoded or baseband, AES, SDI or DVB-ASI, AERO.1000 offers extremely high density. Performing control as a hybrid between single-ended and metadata processing, AERO.1000 preserves quality. Designed and assembled in the USA, the lightweight and rugged AERO.1000 is a solid investment in performance and flexibility. As new processes are discovered, AERO.1000 will be our go-to platform for delivering them to the industry.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. Failover bypass relays on all I/O maintains signal continuity. Dual auto-ranging power supplies for redundant worldwide compatibility.

LOUDNESS

THE LISTEN TEST

Audio loudness processing is becoming an increasingly important part of the broadcaster's job as regulatory imperatives and sophisticated listeners demand more consistent audio levels. Not doing so can risk fines from regulatory agencies and drive viewers to competing providers that have their audio under control.

There are many options when purchasing audio processors from manufacturers worldwide. Any broadcaster that is about to put out their hard-earned money for a processor needs to understand the differences between competing techniques and claims, and most importantly how to evaluate equipment from different manufacturers.

A logical place to start is with an understanding of loudness measurement techniques and (where applicable) the regulations that reference them. The ITU-R BS.1770 measurement standard is the basis for loudness regulation in the US and Europe, as well as other technical regulations. One point of confusion about this standard is that some processing manufacturers claim they are "BS.1770" compliant, even though BS.1770 is a measurement, and not a processing, standard.

One of the key points of BS.1770 is that loudness is a quantity that is integrated over a certain time period. While shorter integration times can be used, it is valid to measure entire programs (with an infinite integration time that is reset between programs). Unlike PPM and VU level meters, instantaneous loudness measurements have little utility.

This is done for a straightforward reason: audio has dynamics. Digital audio broadcasts provide a much wider usable dynamic range than did analogue transmission standards. The average loudness level of audio content can be measured over time using a BS.1770 while still preserving this dynamic range.

Some broadcasters who have evaluated loudness processors are surprised to see the output loudness level moving. They are especially sensitive when there are imperatives that give target loudness levels. Yet these target loudness levels are specified over an entire program or even an entire broadcast day for a TV station. Within any normal program there are going to be variations around the target loudness level. This is to be expected, since the long term or average loudness is what is important.

The type of program content currently being processed will greatly affect dynamic range as well. Certain types of content - for example, some sports with constant crowd noise and near-constant play-by-play announcers - has relatively constant levels. Other types of content, such as classical music has wildly varying dynamics. Program producers edit their program audio with louder and softer portions for dramatic effect. These same producers will often object if the peak to average ratio of their programs is drastically altered during broadcast. Listeners with higher-end audio reproduction systems will also notice more limited dynamics.

So what is best when processing such varying types of content? As a general rule, preserving dynamics is beneficial. Using loudness processors with a range of preset conditions greatly simplifies set up for the broadcaster. Linear Acoustic Audio Loudness Managers have a range of presets that allow adjustment of all processing parameters. Included in this list are presets that minimally alter the peak-to-average ratio of content while still achieving target average loudness values. Of course there are also "denser" presets that have considerably lower peak-to-average ratio.

Both of these (and all of the other processing presets) will produce audio with long term loudness at the target level. Many other devices from other manufacturers will do so as well. So what are the differences between loudness managers from Linear Acoustic and others?

While the very definition of an audio loudness processor requires that the output measure correctly on a meter, it is not nearly enough for that to be the only function. The other critical aspect of a processor is that. . . it actually sounds good while maintaining loudness. No meter yet invented can convey "sounds good".

The ultimate test of any audio processing device is a listening session using the best audio evaluation tool in existence, our human ears. A near infinite list of subtleties exists when processing signal content with the dynamics of broadcast audio. Many manufacturers claim to have mastered these subtleties, but a few listening tests almost always provide a different story. Imagine a ridiculously extreme example: a loudness processor could consist of a gain circuit and a clipper. While it would stick the loudness meter right on the target value, it would be completely unlistenable.

Sadly this technique is not too far off from some other manufacturers' current product offerings. Other products are simplistic wide band gain control circuits that completely ignore the complexities of the human ear. Fletcher and Munson taught us in the 1930s that our human ears are not at all equally sensitive to stimuli at different frequencies, and that this response varies with sound pressure level. Amazingly some designers of audio loudness processors seem completely unaware of this work done almost 80 years ago.

So evaluating a loudness processor actually turns out to be a fairly straightforward process. First, ensure that the candidate processor will produce the target loudness level over a long time frame, realizing that there will be short term variations about the target level due to normal audio dynamics. Almost every existing processor will achieve this. The next step is quite simple but will be the real test: listen to the processed outputs with a wide range of content. Chances are our human ears will be more revealing than the most in depth data sheet.

Mike Richardson
Director of Products

AERO.file

AUDIO/LOUDNESS MANAGER

AERO.file® brings proven audio technologies developed by Linear Acoustic to the file-based domain. Developed in partnership with Radiant Grid Technologies, AERO.file eliminates the need for custom hardware and integrates audio processes into existing systems and workflows.

Tools can be used for ingest, managing existing libraries, conforming content for different playout services, or any combination.

In the TV audio process, upmixing and loudness range control tools have proven most effective in the file domain. Advanced RadiantGrid transmuxing and transwrapping enables the audio essence to be extracted from a host of popular file wrappers, measured, scaled, and processed, then re-wrapped without disturbing other video or data essences.

AERO.file supports WAV, AIFF, MPEG 1 Layer II, MP3, AAC, ACELP, WMA, AMR and uses SurCode for Dolby® Digital, Dolby Digital Plus and Dolby E encoding and decoding.

Operator controls are simplified to choices of loudness target, whether to use 2-channel to 5.1 channel upmixing and whether to use loudness range control.

WHEN SIMPLE SCALING IS NOT ENOUGH

Whether anchor-based such as with Dolby Dialogue Intelligence™ or overall average with the relative gating methods of EBU R128, scaling aligns the anchor or overall average of content. This can easily be imagined when considering how to match a commercial advertisement with a program filled with dialogue and explosions - what do you match with what?

Sometimes programs have a loudness range, that while appropriate for a movie theatre or a premium channel, is challenging to re-purpose for delivery on other channels and especially mobile services.

This is where sophisticated loudness range management techniques can be employed. Once scaling is applied to the program, the job of loudness range control (LRC) is dramatically simplified and the effects are minimized.

AERO.mobile

AUDIO/LOUDNESS MANAGER

AERO.mobile™ is rich audio clarity for Mobile DTV. Part of convincing an audience that Mobile DTV is an exciting option means providing them with an enjoyable experience. Viewers are listening. AERO.mobile is designed to ensure viewers not only hear and understand content, but are surprised by the clarity.

Mobile DTV must overcome physical constraints – small speakers and environmental issues such as background noise. In addition, program audio can range from mono to 5.1 channels and from faint to screaming loud. These factors combine to impair intelligibility, make viewing tedious and cause viewers to give up.

Traditional audio processing alone cannot enable diverse audio content to be reproduced effectively from mobile and handheld devices. In fact, it can make the situation worse. Simple wideband techniques don't do enough, and multiband systems soften important audio cues if overused – both negatively impact intelligibility.

Linear Acoustic Mobilizer™ technology was developed based on extensive research into normal and impaired hearing in both quiet and noisy environments. By using technology from the renowned CrowdControl™ algorithm to isolate dialogue elements and combining new multiband techniques designed to preserve critical audio cues, program intelligibility is enhanced without the need for heavy handed processing. Mobilizer also provides pre-conditioning for the low bit rate HE AAC Mobile DTV audio encoder to maximize its performance at even the lowest rates.

Importantly, Mobilizer technology has been carefully designed and tuned to support and enhance systems like Dolby® Mobile which are being introduced for use within mobile receiving devices.

The rugged 1RU AERO.mobile is intended to be installed directly before the mobile audio encoder in either the AES or SDI paths. Bypass relays are provided to ensure continuous service in the unlikely event of failure.

A bright LED display, rotary encoder, and four control keys provide straightforward menu navigation and function adjustment. Dual, redundant, auto-ranging power supplies are available to allow for trouble-free operation worldwide.

LQ-1000

LOUDNESS QUALITY MONITOR

LQ-1000™ gives vibrant clarity to loudness quality metering. Supporting the latest ITU-R loudness measurement standards, LQ-1000 can also include Dolby® Dialogue Intelligence™.

The LQ-1000 difference is in the display. A colorful long-life OLED groups critical loudness parameters like short, medium and long term loudness, loudness history, current peak level, maximum peak level, and the loudness target.

Color is employed to represent the roughly 16dB wide loudness “comfort zone” which is aligned around the adjustable target level. The visual is simple: blue is too quiet, green is just right, yellow is getting loud, and red is too loud. The large LKFS loudness number also changes color to better indicate if the number matches the chosen target.

The LQ-1000 includes two sets of meters to simultaneously measure a 5.1-channel program and a 2.0-channel program. The second meter can alternatively display an internally created LoRo or LtRt downmix. The meters can also respond to metadata applied as serial data or from the VANC space of an applied HDSDI signal, showing the effects of dialnorm and coding mode. True peak metering is also provided.

Loudness history is an essential part of useful loudness measurement, especially for long form programming. The LQ-1000 loudness histogram allows loudness trends to be easily seen, and immediately highlights problem sections.

The LQ-1000 now provides logging to a network drive, which allows stations to keep a record of their loudness measurements should they need to examine the data for a particular date and time. Any LQ-1000 can be updated via software to incorporate this feature.

Dolby Dialogue Intelligence is available and provides the most accurate estimate of loudness possible in an automatic meter. Pausing integration during non-dialogue sections and reverting to BS.1770-2 over time, loudness can finally be measured with accuracy independent of program dynamic range.

Common functions such as measurement Start, Stop, and Reset are controlled by dedicated front panel buttons - no need to dig through menus. A powerful, high quality 6.3mm (1/4”) headphone output is provided.

A VGA output to feed an external monitor is standard. Options include Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), and Dolby E decoding, and a 7" remote VGA display.

LQ-1

ITU-R BS.1770 LOUDNESS METER

SDI, AES, Analogue Inputs Standard, HE-AAC, MPEG-1, Layer II, Dolby® Decoding

LQ-1™ is perfect streamlined metering. When only one single number is all that is necessary, that number must be correct. LQ-1 manages complex I/O, Dolby decoding, and metadata to provide metering accuracy in a compact, cost-effective package.

» ITU-R BS.1770 Compliant Metering **»** Simple display of signals and loudness **»** HD/SD-SDI, full 16-channel de-embed **»** Discrete AES inputs, downmix AES output **»** HE-AAC, MPEG-1, Layer II, Dolby E/D/DD+ decoding **»** Dolby Dialogue Intelligence™ **»** Stereo analogue input **»** Headphone and +4dBu balanced monitor out **»** Selectable LtRt or LoRo downmix **»** GPI/O Alarms and Control **»** DVB-ASI Input (option) **»** Ethernet for Logging and SNMP (option) **»** Dual PSU (option) **»** Fully upgradable future-ready platform

LQ-1 provides all necessary I/O, routing, decoding and metadata tools.

Input signal levels are displayed alongside the loudness or dialnorm target and measured loudness is continuously displayed. Meter running status (start/stop) is shown as well.

Setup is simple. Select the desired input signal and choose to apply Dolby decoding and metadata if needed. Presets store diverse configurations and can be recalled from the front panel or by GPI.

Extensive alarm capabilities can indicate out of tolerance loudness, missing audio channels, corrupt or missing reference or metadata signals, and errors in Dolby-encoded bitstreams can be detected and logged.

A downmix of the input signal is available as stereo LoRo or surround LtRt, and is provided simultaneously via the front panel 1/4" headphone output, +4dBu balanced analog output and an AES output.

Included Dolby Dialogue Intelligence provides the most accurate estimate of loudness possible in an automatic meter. Pausing integration during non-dialogue sections and reverting to BS.1770-2 over time, loudness can finally be measured with accuracy independent of program dynamic range.

Designed and assembled in the USA, the lightweight and rugged 1RU aluminum chassis is durable enough for installation in challenging environments like OB trucks and cramped edit bays.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. A medical-grade auto-ranging power supply provides trouble-free operation worldwide.

UPMAX

5.1 CHANNEL UPMIXER

UPMAX® delivers smooth transitions. Listening viewers are aware of programming changes, especially when the image shifts due to cases where stereo programs can only be reproduced from the Left and Right channels of a 5.1 channel program. This is commonly found in situations where it is not possible to switch the Dolby® Digital (AC-3) encoder. UPMAX is the simple, well-proven, cost-effective solution.

Based on the original UPMAX 2251, the 1RU UPMAX offers the most stable and trusted algorithm in use today for both production and unattended upmixing. Output is completely downmix compatible and the downmixed result is nearly indistinguishable from the original two channel input.

The resulting "Surroundfield" can be infinitely adjusted via the Center Channel Width control and the Surround Depth control. This allows programming ranging from simple stereo to LtRt to be appropriately reproduced from a 5.1-channel playback system.

An optional bass enhancement signal for the LFE channel is derived from the Left, Center, and Right channels allowing quick creation of a subwoofer channel without compromising the downmix. Factory presets are included for typical applications such as music and commercials. Further adjustment is simple and new results can be stored as user-defined presets.

UPMAX includes a utility encoder which accepts 5.1 channels and produces a two channel LoRo or LtRt output. This encoder can be independent or it can be fed by the same channels applied to the upmixer.

UPMAX is rugged and perfect for remote OB trucks, post production facilities, network operation centers, local station production, virtually anywhere upmixing is used.

Upmixing can be controlled via the front panel, GPI inputs, or metadata from serial or VANC (SDI) sources applied to the unit. Smooth bypass of 5.1 channel signals is accomplished automatically via the AutoMAX-II algorithm.

A bright LED display, rotary encoder, and four control keys provide straightforward menu navigation and function adjustment. Dual, redundant, auto-ranging power supplies are standard to allow for world-wide operation. Bypass relays are provided for trouble-free operation in transmission critical environments.

LA-5269

DOLBY® DIGITAL / PLUS TRANSCODER

Encode, decode, and transcode the most popular multi-channel audio formats used in television broadcasting in one feature-rich, modular package:

Right out of the box, the LA-5269 can: Encode to Dolby Digital (AC-3) from PCM, Encode to Dolby Digital Plus (E-AC-3) from PCM or transcode from Dolby Digital (AC-3), Encode to Dolby Pulse from PCM or transcode from Dolby E.

Optionally, it can: Decode Dolby E and transcode to Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), or Dolby Pulse. Encode to Dolby Pulse (AAC and HE AAC V1 and V2).

All coding is provided by a Dolby-manufactured encoder module featuring the latest versions of each codec for superior sound quality.

Because features and codecs can be updated at any time via software, broadcasters pay for only what they need at the time, knowing that they can always update the unit as their needs change.

Metadata is supported via a serial RS-485 connection and can be extracted from the VANC space of an applied HD-SDI signal per SMPTE 2020. Metadata input is frame synchronized and error-corrected to prevent audible disturbances to the encoded bitstreams. External transcoder input is also frame synchronized, allowing Dolby Digital splicing and smooth transitions without the need for external AC-3 frame synchronizers.

A bright LED display and rotary encoder with four control keys provide easy menu navigation. Dual redundant power supplies, GPI/O, and a hard relay bypass are standard, while SNMP monitoring is offered as an option.

LAMBDA

PROFESSIONAL DIGITAL AUDIO AND METADATA MONITOR

LAMBDA™ is the ultimate digital TV broadcast audio monitor. Designed specifically for the specialized needs of the modern TV station, LAMBDA combines a unique understanding of audio and metadata through the entire broadcast chain from production to consumer.

LAMBDA displays and reproduces up to sixteen audio channels via AES or HD/SD-SDI input, and accepts industry standard professional audio metadata via 9-pin serial input or by extracting it from the vertical ancillary (VANC) space of an applied HD-SDI input. Audio and metadata are displayed and properly combined to allow for accurate monitoring. A utility audio delay is included to allow up to three frames of compensation for external video monitors.

Any channel, channel pair, or downmix can be monitored through internal speakers, via the exceptionally dynamic front panel headphone output, or from the rear panel balanced analogue stereo and AES digital output. A new 16-channel mode allows all applied audio channels to be displayed simultaneously and reproduced individually or as a 5.1 downmix.

High-excursion full range drivers with aluminum cones are coupled with metal dome HF drivers in an acoustically tuned enclosure to optimize frequency response and power handling. Digital Linkwitz-Reilly style crossovers are combined with low distortion, high efficiency class-D power amplifiers for exceptional audio quality and loudness.

Loudness metering per the ITU-R BS.1770 standard is also included. In addition to a numerical readout, a thin line indicating measured loudness "floats" over audio metering to allow quick verification of program loudness.

WHEN BROADCASTERS SPEAK, WE LISTEN

Having the UPMAX with us in Beijing this past summer was like adding a new friend to the crew. The sound was a nice improvement, and the support from Linear Acoustic was superb. We are still learning about new ways to use UPMAX, and I look forward to using it and working with Linear Acoustic again in Vancouver.

Bob Dixon
NBC Universal

Having great 5.1 surround sound accompany HD pictures is a necessity with an event of this magnitude. The Linear Acoustic e-squared system was one of the multichannel audio paths used for distributing the programs for broadcast. This is one of the most watched broadcasts in the world, with entertainment moments that are preserved for history, so audio quality and reliability were critical for us.

Tad Scripter
Engineer in Charge for the 81st Academy Awards

The LAMBDA is a top-shelf piece of gear. It is definitely the future of broadcast facility audio monitoring.

Joey Gill
Chief Engineer
WPSD-TV

KMOV has been using the AERO.air (5.1) for two months and we are very happy with the results. The quality of our audio signal improved noticeably when we placed the unit on the air. The 5.1 channel synthesizing and audio leveling is substantially better than with our previous device. The internal audio/video frame synchronizer function completely cleans up the signal and has corrected a problem with incompatible audio frames on switches. Linear Acoustic did everything possible to ensure that the installation and configuration was correct for our particular needs. I could not be more pleased with the company or the product.

Walt Nichol
Director of Technology, Broadcast Media
KMOV-TV St. Louis

The Linear Acoustic AERO.one is definitely one of the easiest-to-set-up pieces of audio processing gear I have ever experienced. Plus, it sounds great with little or no effort. Having been in the business for 40-plus years, I have seen my share of audio processing and this unit, by far, is my favorite. It "fixes" the levels the network sends us in a very pleasant way and makes the viewers very happy.

Tom Bondurant
Director of Engineering
WAPT-TV

Modern digital broadcast audio such as 5.1 surround sound and its metadata have made QC monitoring extremely important to our operations. In our move to a digital environment, we needed an advanced solution that would appropriately adjust metering and playback audio levels throughout the entire broadcast chain. We chose the Linear Acoustic LAMBDA based on an expectation of excellent audio quality, and that is exactly what we see. We've had the units in place for several months, and they have provided exceptional performance across all three of our channels.

Gene Talley
Director of Engineering/Operations
WPBT-TV

Like many broadcasters, we experienced a lot of problems with varying audio levels for network and local programming, particularly during playout of news and sports. We needed a way to address this discrepancy in loudness levels and even out audio volume, and the Linear Acoustic AERO.air has proven to be a wonderful solution. We noticed a significant difference immediately upon implementing the processor, and we haven't received any comments about disparities in audio loudness since.

Brent Robinson, Chief Engineer
KSL 5-TV

After switching our Comcast channel delivery from analog to digital, we discovered that our audio levels were out of control and we had cracking and popping that we could not resolve, causing viewer complaints every day. We called Linear Acoustic, and they offered to locate and fix the problem for us, leave the equipment in for us to try, and for a very affordable price - a no brainer. Wow! No more complaints from the viewers or the boss - just perfect audio at all times, and in full-time 5.1.

Jan Strock
Director of Engineering
WHTM-TV

Our AERO.air was installed and placed into service this past April. The unit integrated seamlessly with our existing equipment, and I've certainly been impressed with the overall quality of the 5.1 surround sound it provides. If that weren't enough, viewer concerns over commercial loudness have been virtually eliminated and we are now prepared as the CALM Act passed into law.

Moreau Dugas
Engineering Operation Manager
WSVN-TV

Our viewer complaints concerning audio immediately went to ZERO, and we sound great. Not much more to say besides today digital stations just plain need one.

Brady Dreasler
QNI

AERO.max 5.1 is IMPRESSIVE and although I am far passed being able to be impressed, I am with this gadget. The fact that we were able to easily insert our EAS as well was icing on the cake. We put it online with an external Dolby 569 encoder and last night I watched at home with my wife. Wow! The 5.1 is at the output all the time, simulated when local stereo is used, and passed as 5.1 from the network when available. The leveling makes transition between local and network material seamless, and I do not hear (or see) my Onkyo receiver switching surround modes during the breaks either. It has provided WJCT a very uniform and constant off-air sound and fixed dialnorm settings no matter where the material is coming from. FABULOUS gadget.

Duane Smith
Director of Technology
WJCT-TV and FM

Our loudness control problems have virtually ceased thanks to the AERO.air (5.1). The LAMBDA is a very powerful (and cool) box. I still need to teach myself how to use it to its fullest potential and hope to add the AC-3 option later this year which will make it a huge addition to my troubleshooting arsenal.

Prentiss Laird
Engineering Technical Manager
CBS 42 KEYE

We own two AERO.air (DTV) units and are extremely pleased with their performance. Both units were easy to configure and have provided reliable processing and level control. We also selected these units for the ease of 5.1-to-stereo downmix. We have agreed to supply our cable providers a direct SD feed for several more years. This downmix ability provides us with a single platform solution for both our HD and SD feeds. We monitor all feeds with our LAMBDA monitoring unit. It gives us a good handle on our 5.1 processing and the dialnorm of any feed on our wideband router.

Jay Nix
Director of Engineering
KSHB-TV